Toneboosters

TrackEssentials v3
Installation and user manual

Contents

1 Setting up the plugins for first use

1.1 Installation

Download and install the freely-available trial/evaluation versions of the plugins. This will allow you to test the plugins prior to making any purchase decisions. trial/evaluation versions can be downloaded from the ToneBoosters.com website. Trial/evaluation plugins can have one or more of the following limitations:

- Trial/evaluation versions will not store nor save settings.
- Trial/evaluation versions will show a reminder to purchase a license.

1.1.1 Installing the trial/evaluation plugins

- The plugins will be installed in the locations given in the table below.

Operating system	Plugin folder(s)
Windows	`C:\Program Files\Common Files\VST2\ToneBoosters\`
Mac OS	`/Library/Audio/Plug-Ins/Components/` `/Library/Audio/Plug-Ins/VST/`

1.1.2 System requirements (Windows)

- Windows 7 SP1 or higher
- 32 or 64 bit host program that supports VST plugins

1.1.3 System requirements (mac OS)

- macOS 10.10 or higher
- 64-bit host program that supports 64-bit VST or Audio Unit plugins

1.1.4 Configuring your host program

After installation of the trial/evaluation plugins, you may have to inform your host program about the presence of new plugins. Most host programs require you to provide the folder where plugins are installed.

- Consult your host program manual how to configure plugin folders. On Windows, make sure you add the following VST scan path to your host program settings:
  `C:\Program Files\Common Files\VST2\`
- Refresh and/or re-start your host program to allow it to scan for new plugins on your computer.

1.2 Upgrading from trial/evaluation to full/registered versions

Existing installations of trial/evaluation plugins can be upgraded to fully functional versions with a separate registration 'key file'. The registration key file can be acquired in the online shop at www.toneboosters.com. The same key files can be used for Windows and OS X. After purchase:

- Download and extract (unzip) the registration key file(s) ('TB_PluginName.key') and place it in the exact same folder as the corresponding demo/evaluation version of the already installed VST or AU plugin(s).

- On a Windows computer, in one and the same directory, you should see the following pair of files for each registered VST plugin:

  ```
  TB_PluginName_v3.dll
  TB_PluginName.key
  ```

  On a Mac, point finder to your /Library/Audio/Plug-Ins folder. Typically, the Library folder is a 'hidden' folder so you cannot simply browse to that folder. Instead, type Command+Shift+G from the Mac desktop (or Finder > Go > Go to Folder) and type in */Library* to temporarily access the Library directory in the Finder. Then navigate to your VST and Components folders and make sure you copy key files to see the following:

  ```
  TB_PluginName_v3.vst
  TB_PluginName.key
  ```

  ```
  TB_PluginName_v3.component
  TB_PluginName.key
  ```

- Restart the host program. The plugin should now display 'registered' in the lower-right corner of the GUI, instead of 'demo'.

Please make sure to make a backup copy of this registration key file; if the registration key file is lost or damaged the plugin will automatically downgrade to a demo version. Your computer's hard drive is **NOT** a good place for a backup.

Do not rename nor edit the key file. The registration key file comes in a zip archive. Just unzip the archive and copy the resulting key file into your plugin folder. Renaming or modifying the file will cause the registration key file to become dysfunctional.

Some FREE plugins also have an associated registration key file. This registration key file is included in the evaluation download package and allows verification that the key registration system works on your computer. Please do not delete these key files as it will downgrade these free plugins to demo/evaluation versions.

1.3 Installing plugin updates

Download the latest and greatest trial/evaluation versions from the downloads page at www.toneboosters.com and overwrite your plugin files with the newer ones. **Do not delete or modify your registration key files!** The registration key files are the files that have the 'key' extension. Restart your host program and you are all set.

1.4 Disclaimers

VST is a trademark of Steinberg Media Technologies GmbH.

2 User interface common controls

2.1.1 Controlling Knobs and sliders

The various knobs and sliders on the graphical user interfaces (GUIs) of the plugins can be controlled by left-mouse clicks (for switches) or left-mouse drags (for rotary controls and sliders). The following key combinations apply that modify the behavior of the GUI elements:

Windows:

- 'Control' key + left mouse click: set the control at its default value.
- 'Shift' key + left mouse drag: fine-tuning of the control.
- 'Alt' key + left mouse click + mouse move: jump to the clicked position.
- Mouse wheel: change the value up or down.
- 'Shift' key + Mouse wheel: fine-tuning of the control.
- Left or down key: change value down.
- Up or right key: change value up.
- Double left click (if the control has a numeric entry): manual data entry.

OSX:

- 'Command' key + left mouse click: set the control at its default value.
- 'Shift' key + left mouse drag: fine-tuning of the control.
- 'Alt' key + left mouse click + mouse move: jump to the clicked position.
- Mouse wheel: change the value up or down.
- 'Shift' key + Mouse wheel: fine-tuning of the control.
- Left or down key: change value down.
- Up or right key: change value up.
- Double left click (if the control has a numeric entry): manual data entry.

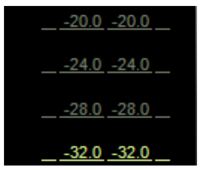
2.1.2 Controlling nodes

Some plugins have nodes (round dots) in a two-dimensional space or graph that can be dragged to change parameters. These can be dragged with the mouse to change their position and the associated parameter values. The following key combinations apply:

- Left mouse click: activate node.
- Right mouse click: de-activate node.
- Drag with left-mouse knob: modify position in active mode.
- Drag with right-mouse knob: modify position in inactive mode.
- Drag with left-mouse knob and 'alt' key: lock Y while moving node
- Drag with left-mouse knob and 'control'/'command' key: lock X while moving
- 'Shift' key + drag with left-mouse knob: fine-tune position in active mode.
- 'Shift' key + drag with right-mouse knob: fine-tune position in inactive mode.
- Mouse wheel: change secondary parameter (for example filter quality).
- 'Shift' key + mouse wheel: fine-tuning of secondary parameter.

2.1.3 VU meters

VU meters will often support 'peak hold' functionality, in which the most extreme value across time is indicated by a horizontal line with the peak value displayed numerically above this line.



- Click on the VU meter to reset the peak hold value (if supported).
- Drag the VU meter scale to change its range (only in a limited set of plugins)

2.1.4 Bypass function

Most ToneBoosters plugins do not have a bypass switch – since virtually all VST hosts have their own bypass functions it is assumed that bypass operation is managed by the host, not by the plugin.

3 TB Ferox v3

3.1 Introduction

TB Ferox provides smooth compression and saturation that reminds one of the good old days of tape recording, and allows for additional feedback-delay effects, which makes it especially suitable to artistically enhance and enrich individual audio tracks. TB Ferox is a general-purpose tape simulation plugin that differs from TB ReelBus in the following ways:

- TB ReelBus is accurately modeled after tape recorders; TB Ferox consists of a more general and simpler tape saturation simulation.
- TB ReelBus has tape hiss and asperity noise models, these are not available in TB Ferox.
- TB ReelBus has a wide variety of tape models; TB Ferox has saturation and hysteresis controls that can work in 2 modes.
- Besides non-linear behavior, TB ReelBus also models certain spectral features of tape, such as the head bump.
- TB ReelBus has configurable pre- and post-emphasis, which are not available in TB Ferox.
- TB ReelBus has wow and flutter simulation, this is not present in TB Ferox.

In general, TB Ferox has a lower CPU load and is therefore very suitable for individual tracks, while TB ReelBus fits better with busses and complete mixes (although it can sound pretty good on individual tracks as well).

3.2 User interface

GUI Control	Purpose
Drive	Sets the input 'drive' level for the plugin. A higher value will push the signal harder into saturation and compression, giving a more pronounced effect. The effective signal level is visualized by the VU meters.
Compress	Sets the amount of tape compression that is applied. A value of 0% will disable tape compression.

Release	Sets the release time constant associated with tape compression. A higher value will result in a slower release time.
Saturate	Sets the amount of tape saturation that is applied. A higher value will result in more saturation, and therefore increase the level of tape harmonics being produced by the saturation process.
Hyst	Sets the amount of hysteresis that is applied. A higher value will result in stronger hysteresis effects. Hysteresis effects are mostly audible with content comprising energy at high frequencies, such as percussion, hihats and alike. Because hysteresis is associated with processing high frequencies, it is advised to use high values of hysteresis in conjunction with enabling oversampling to prevent aliasing distortion.
Mode	Toggles between two models for tape non-linearity. The ‘low’ setting will typically give a softer, less pronounced effect than the ‘high’ setting.
Oversample	Enables / disables internal oversampling. It is recommended to set this to enabled if significant amounts of saturation and/or hysteresis are being applied to prevent aliasing distortion.
Out gain	Sets the output gain in decibels.
Power	Disables / enables the plugin. When disabled, the input signal is fed to the output without any processing (bypass).
Feedback	Sets the amount of output being fed back into the input. This feature is especially useful for tape echo simulation.
Delay	Sets the duration of the delay of the feedback path. Change this value to change the delay time for tape echo simulation.
HP frequency	Sets the high-pass frequency applied during the tape simulation algorithm (requires the filter control to be set to ‘on’).
LP frequency	Sets the low-pass frequency applied during the tape simulation algorithm (requires the filter control to be set to ‘on’).
Mono	Enables / disables the mono operation mode.

4 TB Module v3

4.1 Introduction

TB Module is a generic framework plugin for a wide variety of feedback, non-linearities and modulation effects. It consists of a set of serialized processes such as panners, delays, filters, decimators, quantizers, and sound-field rotation modules with user-controllable feedback. The saturation effect is ported from TB Ferox, and the quantizer/decimator is ported from the acclaimed TB TimeMachine plugin. In combination with 2 LFOs, effects such as chorus, tremolo, auto pan, phaser and tempo-synced filter sweeps are created easily. Moreover, because of its generic approach, TB Module allows for hybrid combination effects as well.

4.2 User interface

GUI section	Control	Purpose
Filter	HPF	Sets the high-pass cut-off frequency in Hz. Use this to remove low frequencies from the output.
	LPF	Sets the low-pass cut-off frequency in Hz. Use this to remove high frequencies from the output. The control below the cut-off frequency sets the amount of resonance of the low-pass filter.
Non-linearities	Compress	Sets the amount of tape compression that is applied. A value of 0% will disable tape compression.
	Saturate	Sets the amount of tape saturation that is applied. A higher value will result in more saturation, and therefore increase the level of tape harmonics being produced by the saturation process.
	Decimate	Sets the decimation (resampling) factor. A value of *x* will simulate down-sampling by a factor of *x*, without anti-aliasing filters for creative purposes.
	Quantize	Sets the amount quantization. A higher value will simulate the effect of a D/A converter with less bits, resulting in a grainy, noisy character.
Delay	Amount	Sets the duration of the delay of the feedback path. Change this value to change the delay time for echo simulation. The delay can be specified in seconds, or as number of beats used by the host for tempo-synced effects (if supported by the host).
	Feedback	Sets the amount of output being fed back into the input. This feature is especially useful for echo simulation and phaser processing. A positive value will result in

		positive feedback; a negative value produces negative feedback (e.g., the output signal being fed to the input including a 180-degree phase shift).
	Oversampling	Enables / disables oversampling.
Mixer	InPan	Changes the stereo panning of the input before processing.
	Rotate	Sets the amount of stereo rotation that is applied every time the signal pass through the plugin. When feedback is enabled, this allows the stereo image to rotate by a certain amount for every echo / repetition.
	OutGain	Sets the gain of the processed signal.
	Wet out	Sets the ratio of processed and unprocessed input produced at the output of the plugin.
LFO 1/2	Freq	Sets the LFO frequency and waveform. The frequency can be set in Hz, or in beats for tempo-synced effects (if supported by the host). The LFO waveform can be selected from ‘sine’, ‘cos’, ‘square’, ‘saw’. The LFO waveform cycle is visualized by the LED just above the frequency control. The waveform of the LFO is positive when the LED is fully lit; the waveform of the LFO is negative if the LED is off.
	Amount: decimate	Sets how much the decimate control is modified by the LFO. A positive value of the ‘amount’ control will increase the ‘decimate’ setting if the LFO waveform is positive and vice versa.
	Amount: Quantize	Sets how much the quantize control is modified by the LFO. A positive value of the ‘amount’ control will increase the ‘quantize’ setting if the LFO waveform is positive and vice versa.
	Amount: Delay	Sets how much the delay is modified by the LFO. A positive value of the ‘amount’ control will result in *a common* delay applied to left and right. A negative value will result in a delay *difference* between left and right for stereo image manipulation, stereo pitch manipulation, and stereo phaser effects.
	Amount: Outgain	Sets how much the OutGain is modified by the LFO. A positive value of the ‘amount’ control will result in *a common* gain applied to left and right. A negative value will result in a gain *difference* between left and right for stereo image manipulation, such as used in auto panners.
	Amount: LPF	Sets how much the LPF (low-pass filter) cut-off frequency is modified by the LFO. A positive value will result in an increase in the LPF frequency if the LFO waveform is positive and vice versa.

5 TB Gate v3

5.1 Introduction

TB Gate is a gating processor suitable for a range of processes, including background noise gating, creating gated reverb, and alike. The gate has three states:

- ‘Open’, indicating that the input signal is above a minimum threshold level, and the input signal is fed to the plugin output;
- ‘Hold’, indicating that the input signal is low (below the minimum threshold level) and that the gate is about to close.
- ‘Closed’, indicating that the input signal is low (below the minimum threshold level) and that the gate is closed.

The plugin has a dedicated internal side chain to detect the input signal level including high-pass and low-pass filters to adjust the sensitivity of the gate. This internal side chain is referred to as the (input level) detector.

5.2 User interface

GUI section	Control	Purpose
Detector	HP freq	Sets the input level detector high-pass filter frequency. Increasing this frequency will reduce low frequencies from the input level detector, resulting in a reduced sensitivity for the gate to open with low-frequency input signals. The output signal itself is not processed by this filter.
	LP freq	Sets the input level detector low-pass filter frequency. Decreasing this frequency will reduce high frequencies from the input level detector, resulting in a reduced sensitivity for the gate to open with high-frequency input signals. The output signal itself is not processed by this filter.
	Detection time	Sets the detection time used by the gate to determine the input level. A longer detection time will result in better (more consistent) handling of noisy signals at the expense of a more sluggish response.
	Threshold	Sets the gate threshold. If the input signal (after applying the input level detector high-pass and low-pass filters) is above this threshold, the gate will open and less the signal pass through.
	Filter on	Enables / disables the input level detector high-pass and low-pass filters.
	Monitor	If enabled, the output of the plugin will consist of the signal used by the input level detector, including the input level detector high-pass and low-pass filters if these filters are enabled.
Envelope	Attack	Sets the time it takes for the gate to transition from ‘closed’ to ‘open’ for smooth fading of the input signal.

	Hold	Sets the minimum time that the gate will remain open. If the input signal level drops below the threshold value, the gate will remain open for a duration set by this control before it transitions into the ‘closed’ state.
	Release	Sets the time it takes for the gate to fade the signal when the state is ‘closed’. During this transition, the input signal will be faded out. A longer release time will cause a slower fade out.
	Close gain	Sets the gain / attenuation of the gate when in ‘closed’ state. A setting of zero will result in a quiet output when the gate is in ‘closed’ state.
Input	VU meter	Provides a visual indication of the input signal level as determined by the input level detector. Use this meter to set the threshold value.
	State	Gives a visual indication of the gate state. • The left (green) LED is lit when the gate is ‘open’. • The middle (orange) LED is lit when the gate is in ‘hold’ state. • The right (red) LED is lit when the gate is in ‘closed’ state.

6 TB DeEsser v3

6.1 Introduction

TB DeEsser removes or reduces excess sibilance in vocals (such as ‘sss’ and ‘tth’) without adversely affecting the low-frequency (voiced) parts of the signals. A variable cross-over frequency and de-essing threshold allow for a precise control of the de-essing function. TB DeEsser’s stereo, split-band design also allows for limiting of any excess high-frequency content in other signals including full stereo mixes.

6.2 User interface

GUI section	Control	Purpose
VU meters	Sibilance	Gives a visual indication of the detected sibilance level in dB for the two input channels.
	De-essing	Gives a visual indication of the reduction in excess sibilance in dB.
Controls	Attack	Sets the attack time, which is the time for the de-esser to react to excess sibilance.
	Release	Sets the release time, which is the time for the de-esser to recover from excess sibilance.
	HP frequency	Sets the high-pass frequency of the de-esser. Excess sibilance detection and removal will start at this frequency.
	Sensitivity	Sets the sensitivity to excess sibilance. A larger value will increase the sensitivity to excess sibilance.
	Amount	Sets the amount of excess sibilance removal. A larger value will result in more aggressive excess sibilance removal.
	Stereo link	Sets the amount of linking for excess sibilance removal in the two channels. A value of 0% will result in independent de-essing in left and right; a value of 100% will give identical de-essing in left and right.
	Listen	If enabled, the output of the plugin will consist of the removed sibilance only for monitoring purposes.
	2band	Sets the split-band mode. If enabled, the de-esser will attenuate excess sibilance for frequencies above the value set by the HP frequency only. If disabled, the de-esser will apply a broad-band attenuation of the signal to reduce excess sibilance.

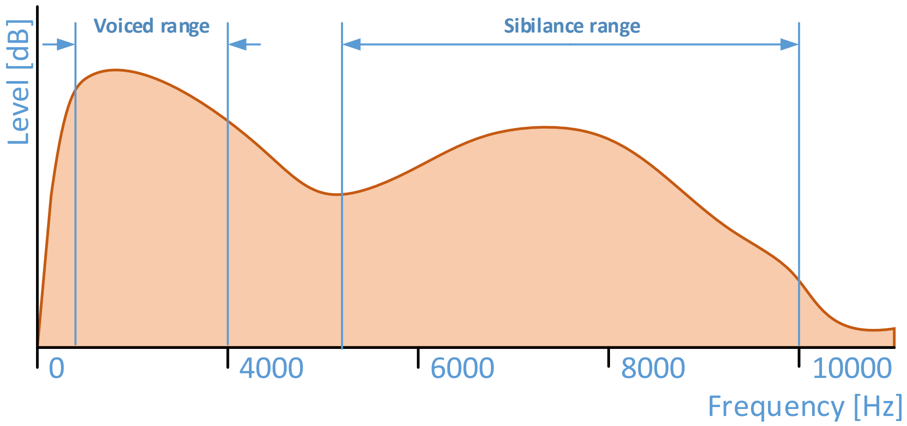
6.3 Sibilance

6.3.1 Voiced and sibilance frequency ranges

The goal of TB DeEsser is to reduce or remove excess sibilance, or said differently, sibilant sounds such as 'ess' that are too loud are to be reduced in level. It is important to realize that the phrase 'too loud', or excess sibilance, is defined within its context. This context dependency is explained schematically below.

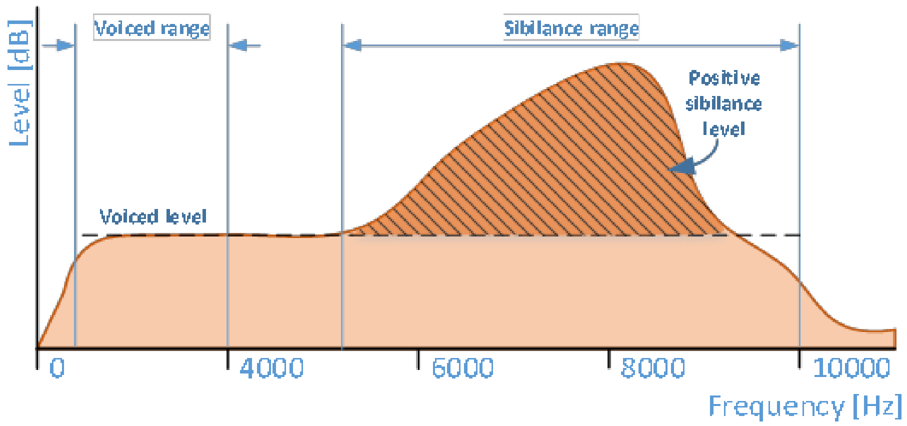
Let us start with showing a spectrum of an audio signal. In the figure below you will see the power spectrum level of a sound as a function of frequency. We can identify two frequency ranges that are not necessarily mutually exclusive (they are allowed to overlap in frequency):

- A voiced frequency range, typically around 200 – 4000 Hz, which is the frequency range in which voiced parts of speech (such as 'a', 'e', 'i', and alike) are predominantly present, and
- A sibilance frequency range, typically around 5000-11000 Hz, which is the frequency range in which sibilant sounds (such as 's', 't', and alike), and excess sibilance often occurs.



6.3.2 Sibilance level

Let us consider an example in which excess sibilance occurs. The figure below one can clearly observe that the power spectrum level within the sibilance range is much higher than the (average) power spectrum level in the voiced range. Schematically, we could therefore consider the power in the sibilance range above the voiced level as *excess* sibilance, as it clearly stands out in the context of the overall power spectrum, and with respect to the voiced level. Said differently, the sibilance level is *positive*. If, on the other hand, the power spectrum level in the sibilance range would be below the voiced level, the sibilance level is *negative*. The sibilance level is visualized in the GUI in real time by the two left-most VU meters.



6.3.3 De-essing amount

The amount control determines how much of the excess sibilance will be removed. When set to 50%, as an example, 50% of the detected sibilance level will be removed. In other words, if the sibilance level indicates a value of +10 dB, and the amount control is set to 50%, the sibilance range will be attenuated by up to 5 dB.

6.3.4 Attack and release

For certain languages, the onset of sibilant sounds is very important for language intelligibility. If a pass-through of such onsets is desirable, and de-essing should only start a short time after such onset, increase the attack parameter. This will cause the de-essing to kick in later and allow pass-through of onsets of sibilant sounds.

Similarly, the release parameter determines how fast the de-esser recovers from attenuating sibilant sounds. A longer value tends to give a smoother behavior, but a too slow value may cause the de-esser to recover too slowly before a non-sibilant syllable starts. For very fast talkers, a shorter release time may therefore be beneficial.

6.3.5 Single vs dual-band operation

When TB DeEsser is running in single band mode, the attenuation to remove excess sibilance is applied to the full frequency range, essentially like lowering the overall volume for a very short moment in time. Therefore, voiced signal elements will be attenuated as well, which may not always be desirable. To overcome this, the dual-band operation mode will only attenuate frequencies in the sibilance range, and leave lower frequencies untouched.

7 TB Compressor v3

7.1 Introduction

TB Compressor is a versatile dynamic range compressor plugin that is suitable for virtually all audio content. Its advanced signal analysis algorithms allow for a very transparent dynamic range modification. Conversely, with manual adjustment of its parameters, the compressor can provide that 'musical pumping effect' to spice up your beats. A dedicated dry/wet control provides a hassle-free New York style / parallel compression.

7.2 User interface

Control	Purpose
Threshold	Sets the threshold input signal level at which the compressor starts to compress (attenuate) the input signal.
Ratio	Sets the ratio (amount) of compression. A ratio of 4 indicates that a signal level of 4dB above threshold will be reduced to 1dB above threshold.
Makeup	Sets the output (makeup) gain in decibels.
Range	Sets the maximum compression amount (range). If the range is set to 20 dB, the compressor will never modify the signal level by more than 20 dB.
%Wet out	Sets the percentage of dry and wet mixing in the output. A value of 50% will mix the (unmodified) input and the (compressed) output with a 50/50 ratio. This control can be used for parallel compression. The effective input/output curve is visualized in the graph.
Sidechain HPF	Sets the high-pass filter cut-off frequency for the internal level detector (also referred to as sidechain). If the control is set to 100 Hz, input signal frequencies below 100 Hz will have a reduced effect on the detected input level, and therefore reduce the amount of compression.

Attack	Sets the time constant for the compressor to react to (sudden) increases in the input signal level. A shorter attack time will result in a faster response, and therefore a more 'snappy' sound.
Release	Sets the time constant for the compressor to recover from high input levels. A longer release time will result in a slower response, and therefore a cleaner (but quieter) sound.
Auto	Enables the auto-release function. If enabled, the release control becomes inactive, and the release time constant is automatically adapted according to the input signal level characteristics (program dependent).
Input/output graph	Gives a visual indication of the input/output curve in decibels. The x axis denotes the input signal level; the y axis gives the corresponding output level for the current compressor settings. • The threshold is indicated by the point at which the curve changes its slope. • The ratio is indicated by the slope of the curve for signal levels above the threshold. • The current input level is shown as a dynamically-changing fill under the input/output curve.
Compression VU	The VU meter next to the input/output graph shows the compressor attenuation in real time, including a peak-hold function indicating the maximum attenuation over time. • Click on the VU meter to reset the peak-hold value. • The overall scale of the VU meter is linked to the makeup gain control. Dragging the VU meter up/down changes the makeup gain. • The maximum of the VU meter scale will always be equal to the value of the makeup gain. As a result, the VU meter will display the combination of (1) the compressor attenuation, and (2) the makeup gain.
Compressor modes	Sets the compressor mode / algorithm. The compressor has 4 different modes / algorithms: 1. **Vintage punch**. In this mode, the compressor emulates vintage compressors with envelope followers operating on the *compressor gain*. In this mode, the compressor gives a punchy sound, and its compression operation on the signal has lots of character and is often clearly audible. 2. **Modern punch**. In this mode, the compressor emulates vintage compressors with envelope followers operating on the internal-side chain *level detector*. This mode typically gives a very punchy, but cleaner compression operation than the 'vintage' punch setting. 3. **Smooth rider**. In this mode, the compressor engages an algorithm that gives the smoothest, cleanest compression behavior using advanced signal and gain riding functions. Use this mode if clean and transparent compression is required. 4. **Pump and breathe**. This mode is specifically designed to create pumping and breathing effects, especially for input signals with a large dynamic range, as is often used in electronic music. Setting the side-chain HPF to a low cut-off frequency will often increase the pumping effect.

8 TB Reverb v3

8.1 Introduction

TB Reverb is a versatile room acoustic simulation and reverberation algorithm. It provides ample controls to change the simulated room, including frequency-dependent reverberation times, room size and shape, variable warmth and cutoff frequencies, and wall diffusion.

8.2 User interface

GUI section	Control	Purpose
Reverb tail	T60	Sets the reverberation time. T60 indicates the time required for the reverb tail to have reduced in level by 60 dB. A longer T60 will thus result in a longer reverberation time.
	Damp freq	Sets the (damping) frequency at which the T60 reverberation time becomes shorter. Below this frequency, the reverberation time will be equal to the time set by the T60 control. Above this frequency, T60 will gradually decrease, as it often happens in real acoustic environments.
	Character	Determines the decrease of T60 with frequency above the damping frequency. A high value will result in longer T60 reverberation time at high frequencies, and therefore a brighter character of the reverberation tail. A small value will result in a more aggressive reduction in T60 at high frequencies, and therefore a warmer sound.
	Mod	Enables / disables modulation in the reverberation tail.
	HQ	Enables / disables the 'high quality' (HQ) mode. In HQ mode, the density of the reverberation tail is higher, creating a more dense and lush reverb, at the expense of a higher CPU load.
Room geometry	Shape	Determines the shape of the simulated acoustic environment. A high value simulates more diversity in the room dimensions; while a low value simulates a shape that is closer to a box or rectangular shape.
	Size	Determines the size (volume) of the simulated acoustic environment. In physical environments, a larger volume often corresponds to a longer T60. For a natural sounding reverb, a long T60 is therefore recommended in combination with a large room size.
	Diffusion	Sets the amount of diffusion when reflections bounce of walls. A low value will create specular reflections, and results in a sparser reverberation tail. A high value will create diffuse reflection patterns, and a more dense reverberation tail.

	Predelay	Sets the (pre) delay applied to the reverberation tail (in seconds). A higher value causes a reverberation tail to be delayed with respect to the input.
	Width	Sets the stereo width of the reverberation. A value of 0% will create a mono reverberation effect; a value of 100% results in a spatially wide reverberation.
	ER level	Sets the level of the early reflections.
Mixer	% wet out	Sets the ratio of input and reverberation in the output of the plugin. A value of 100% will only produce reverberation, and is the recommended setting if the plugin is configured as a send effect bus.
	High pass	Sets the cut-off frequency of the reverberation high-pass filter.
	Low pass	Sets the cut-off frequency of the reverberation low-pass filter.

9 TB Equalizer v3

9.1 Introduction

TB Equalizer is a versatile equalizer with a wide variety of filter prototypes and channel modes. Each of the 6 filter sections can be configured to run as low cut, low shelf, digital bell shape, analog bell shape, high shelf, notch and high cut filter with variable filter gain, frequency and quality. On top of these standard filter types, Pultec-style low and high shelf filters are supplied as well.

Each filter section can also be configured to run in stereo, left only, right only, mid only, or side only. The overall equalizer curve can be morphed between 100% and -100% continuously.

TB Equalizer has an integrated spectrum analyzer that can display overall power spectrum, left/right pan, and spatial ‘width’ expressed as the ratio of side over mid signal.

9.2 User interface

GUI section	Control	Purpose
Global settings	Output gain	Sets the gain applied to the output in decibels.
	Gain effect	Sets the amount of equalization that is applied. A value of 100% will result in 100% of the gain/attenuation of each section to be applied to the signal. The value can also be negative to invert the effect of each equalization section. • Filter sections configured as low-cut or high-cut will not be affected by the value of this control.
Spectrum analyzer	Resolution	Sets the frequency resolution (in octaves) of the frequency analyzer.
	Speed	Sets the speed (integration) of the frequency analyzer.
	Mode	Sets the mode of the frequency analyzer.

		• 'Off': the frequency analyzer is disabled. • 'L+R sum': the frequency analyzer visualizes the sum of the power of the input channels. • 'L/R pan': the frequency analyzer visualizes the ratio of the power of the input channels. A positive value indicates that the left channel contains more energy at the corresponding frequency than the right channel. • 'S/M width': the frequency analyzer visualizes the ratio of the power of the side signal (L-R) and the mid signal (L+R). A positive value indicates that the side signal is stronger than the mid signal.
Section editor	Section handles and editor	The editor will display the frequency characteristics of each of the equalizer sections. Furthermore, when hovering over the graph, a MIDI note is indicated corresponding to the frequency at which the mouse is located. • Drag the section handles 1-6 to change the frequency and gain for each section. • Left or right clicking on the handle enables / disables the section. • Use the mouse wheel to change the section's Q factor value.
	Section	Sets the equalizer section that is currently being edited.
	Filter type	Sets the filter type of the section that is currently being edited. • Low cut: 2nd-order analog high-pass filter with variable resonance. The gain parameter is ineffective. • Low shelf: 2nd-order analog low-shelving filter with variable resonance. • Digital bell: bell-type, peaking filter; digital style. • Analog bell: bell-type, peaking filter, analog filter type. • High shelf: 2nd-order analog high-shelving filter with variable resonance. • High cut: 2nd-order analog low-pass filter with variable resonance. The gain parameter is ineffective. • Low cut + gain: 2nd-order analog high-pass filter with variable resonance. The gain parameter set the overall output gain. • High cut + gain: 2nd-order analog low-pass filter with variable resonance. The gain parameter set the overall output gain. • Bandpass+gain: band-pass filter with variable bandwidth and output gain. • Notch+gain: notch filter with variable bandwidth and output gain. • Low-shelf 2: Pultec-style low-shelving filter with variable resonance. • Digital bell 2: bell-type, peaking filter that has a different shape with positive and negative gains. • High-shelf 2: Pultec-style high-shelving filter with variable resonance. • Low cut 1: 1st-order high-pass filter without resonance. • High cut 1: 1st-order low-pass filter without resonance.
	Channel	Sets the channel(s) the current section is applied to: • 'All': all channels will be processed. • 'Mid': only the mid channel will be processed. • 'Side': only the side channel will be processed. • 'Left': only the left channel will be processed. • 'Right': only the right channel will be processed.
	Makeup	Enables / disables the auto-makeup gain feature. This feature will attempt to keep the loudness of the signals constant with a change of the filter settings by applying an automatically-adjusted output gain.
	Frequency	Sets the frequency parameter of the current section.
	Gain	Sets the gain parameter of the current section.
	Q factor	Sets the quality (Q) factor of the current section.

10 TB TimeMachine v3

10.1 Introduction

TB TimeMachine simulates the effect of vintage samplers and vintage digital gear and includes:

- Sample-rate dependent simulation of aliasing (for A/D and D/A converters);
- Analog circuitry saturation;
- Quantization.

10.2 The user interface

Control	Purpose
Samplerate	Sets the sample rate used for simulating vintage digital gear. A lower value will create a stronger effect.
High pass	Sets the high-pass filter cut-off frequency (in Hz).
AD aliasing	Sets the amount of analog-to-digital converter aliasing simulation.
Saturate drive	Sets the drive value for analog circuitry saturation simulation. A higher value will drive the input signals more strongly into analog saturation.
Number of bits	Sets the number of bits used in the digital quantization simulation.
Dithering	Enables / disables dithering with the digital quantization simulation.
muLaw	Enables muLaw quantization. When disabled, linear quantization is used.
DA aliasing	Sets the amount of digital-to-analog converter aliasing simulation.
Saturate mix	Sets the strength of analog circuitry saturation effect.